



H.323/SIP PoE IP Phone
VIP-103PT / VIP-153PT / VIP-153T
User's Manual

Copyright

Copyright (C) 2005 PLANET Technology Corp. All rights reserved.

The products and programs described in this User's Manual are licensed products of PLANET Technology, This User's Manual contains proprietary information protected by copyright, and this User's Manual and all accompanying hardware, software, and documentation are copyrighted.

No part of this User's Manual may be copied, photocopied, reproduced, translated, or reduced to any electronic medium or machine-readable form by any means by electronic or mechanical. Including photocopying, recording, or information storage and retrieval systems, for any purpose other than the purchaser's personal use, and without the prior express written permission of PLANET Technology.

Disclaimer

PLANET Technology does not warrant that the hardware will work properly in all environments and applications, and makes no warranty and representation, either implied or expressed, with respect to the quality, performance, merchantability, or fitness for a particular purpose.

PLANET has made every effort to ensure that this User's Manual is accurate; PLANET disclaims liability for any inaccuracies or omissions that may have occurred.

Information in this User's Manual is subject to change without notice and does not represent a commitment on the part of PLANET. PLANET assumes no responsibility for any inaccuracies that may be contained in this User's Manual. PLANET makes no commitment to update or keep current the information in this User's Manual, and reserves the right to make improvements to this User's Manual and/or to the products described in this User's Manual, at any time without notice.

If you find information in this manual that is incorrect, misleading, or incomplete, we would appreciate your comments and suggestions.

CE mark Warning

The is a class B device, In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

WEEE Warning



To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste and have to collect such WEEE separately.

Trademarks

The PLANET logo is a trademark of PLANET Technology. This documentation may refer to numerous hardware and software products by their trade names. In most, if not all cases, their respective companies claim these designations as trademarks or registered trademarks.

Revision

User's Manual for PLANET PoE IP phone:

Model: VIP-103PT/VIP-153T/VIP-153PT

Rev: 1.1 (2005, September)

Part No. EM-VIP153TV1

Table of Contents

Chapter 1	4
Overview of the PoE IP Phone.....	4
VIP/-103PT / VIP-153PT / VIP-153T Features	4
1.1 Hardware Overview	5
1.1.1 Front View and Keypad function.....	5
1.1.2 Rear View.....	7
Chapter 2	8
Configuring the PoE IP Phone through LCD Phone menus.....	8
2.1 Initialize VIP103PT/VIP-153PT/VIP-153T	8
2.2 LCD Menu Configuration.....	9
Chapter 3	19
Configuring the VIP-103PT/VIP-153PT/ VIP-153T through Web Pages.....	19
Step 1. Browse the IP Address of your IP phone	19
Step 2. Input the login name and password	19
Step 3. Enter the web interface main screen	19
1. Overview.....	19
Step 4. Start machine configurations	20
2. Network Configuration	20
3. H.323 Configuration (VIP-103PT).....	23
3. SIP Configuration (VIP-153PT/VIP-153T).....	25
4. Phone Configuration (VIP-103PT).....	27
4. Phone Configuration (VIP-153PT/VIP-153T).....	29
5. System Configuration	30
6. Speed Dial.....	32
7. Phone Book.....	32
Chapter 4	33
Voice communication samples.....	33
Peer to peer (P2P) mode	33
Voice communication via SIP proxy server – SIP-50.....	38
Voice communication via IP PBX system – IPX-2000 (Auto-config)	40
Appendix A.....	43
PoE IP Phone Specifications.....	43

Chapter 1

Overview of the PoE IP Phone

Meeting the next-generation Internet telephony service demands, PLANET Technology provides feature-rich, toll-quality Internet telephony service solutions. With 802.3af Power over Ethernet (PoE) LAN interface supported, the VIP-103PT is H.323 v2/v3/v4 compliant with H.235 security support, VIP-153PT is SIP 2.0 (RFC3261) compliant with SIP digest authentication supports. And the VIP-153 is the cost-effective SIP IP Phone.

The VIP-103PT / VIP-153PT / VIP-153T ("IP Phone" in the following term) features high-quality speakerphone technology, and includes an easy-to-use speaker on/off button and call hold/transfer buttons for various voice services.

The IP Phone has additional features such as built-in PPPoE/DHCP clients, password-protected machine management, LCD menu display, speed-dial keys, hands-free speakerphone, last number redial, incoming message indicator, and user-intuitive web administration system.

The IP Phone is self-contained, service-integrated, intelligent phone features offering, and powerful voice processing power. The IP Phone can effortlessly deliver toll voice quality equivalent to the regular PSTN connections utilizing cutting-edge Quality of Service, echo cancellation, comfort noise generation (CNG) and voice compensation technology. Meanwhile, the dual Ethernet interfaces on the IP Phone allow users to install in an existing network location without interfering with desktop PC network connections. When installing the VIP-103PT / VIP-153PT / VIP-153T, SIP IP Phone with IPX-2000 (PLANET IP PBX system), the VIP-153 series IP phones can be easily integrated in your office; via the auto-config support for IPX-2000. No expertise required building up the VoIP network deployment.

Besides, the VIP-103PT / VIP-153PT deliver more convenience, efficiency, innovation and benefits to VoIP in your daily life, the PoE interface and IP PBX shall replace your existing PBX system with more feature and saving more. The IP Phones are ideal solution for office / home use as well as installation for Internet Telephony Service Provider (ITSP) from leading vendors. It's the delivery platform for IP voice services that makes benefit from the VoIP technologies in your daily life.

VIP/-103PT / VIP-153PT / VIP-153T Features

Simple Installation and administration

Configuration of the IP Phone can be performed in minutes via the keypad, or web interfaces. Using the built-in LCD display, the IP Phone offers user-friendly configuration guidelines, machine operation status, call status displays, and incoming call identification.

IP PBX system integration

Via auto-config support for IPX-2000, no expertise required to establish your office voice network. VIP-153T and the VIP-153PT can help you to complete VoIP network deployment in minutes. The same as H.323 model, VIP-103PT is able to be integrated with PLANET IP PBX system IPX-1000/IPX-1500 as one of the extensions in your office network.

Feature-rich keypad IP Phone

The IP Phone integrates a high-quality speakerphone with the Call Hold, Forward and Transfer functions and also provides advanced telephone features, such as 10 speed-dial keys, last number redial, incoming call history indicator in a much more convenient and functional manner than traditional telephone sets.

Dynamic IP address assignment, and voice communication

The IP Phone can act as a PPPoE/DHCP client, automatically obtaining an IP address for Internet access.

Various field applications compliant

The IP Phone is capable of handling both peer-to-peer and H.323GK / SIP proxy registration, authentication to interact with major H.323/SIP gateway/IP Phone in the market. The IP Phone offers the most flexibility and interoperability with PLANET and 3rd party VoIP vendors, allowing the deployment of both simple and complex VoIP networks such as ITSP, PC-to-Phone/Phone-to-PC or enterprise VoIP environments.

Standards compliant

The VIP-103PT complies with H.323 version 4, and supports H.323 fast start, H.245 tunneling, GateKeeper H.235 security mode for different occasions. The VIP-153T / VIP-153PT complies with SIP 2.0 (RFC3261), interoperates with 3rd party SIP voice gateways/terminal/software as well as other PLANET VoIP products. Supported Voice codecs and VoIP technologies are: G.723, G.729ab, G.711u-law/a-law; Voice Activity Detection (VAD), and the Comfort Noise Generation (CNG).

1.1 Hardware Overview

1.1.1 Front View and Keypad function

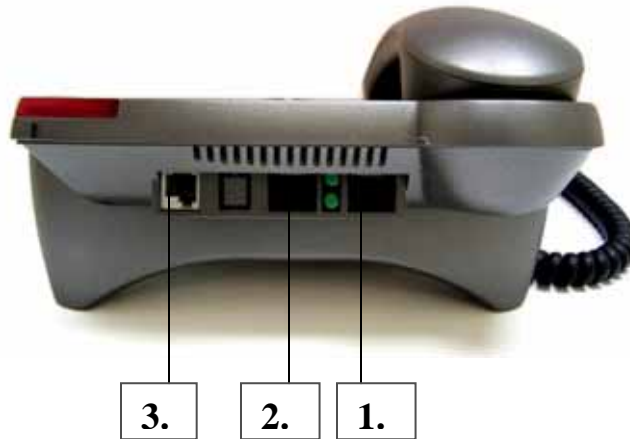


PoE IP phone overview

Keypad Description

1	Light	The red light goes on-off when there is an incoming call.
2	LCD Display	Menu and all status shall be displayed for users.
3	“← UP” Key	To scroll up the menu items or left delete a digit per each key press while editing the settings.
3	“→ DOWN” Key	To scroll up the menu items or left delete a digit per each key press while editing the settings.
4	- IN VOL+	When the IP phone is at idle state or talking on handset or speaker, this key is used to increase/decrease the volume of the voice sound.
5	+OUT VOL-	Users are able to increase or decrease the volume which they send out to remote party.
6	SPEAKER	To switch between the usage of the handset and the speaker devices.
7	HOLD	To hold the conversation.
8	NET	When users are not successfully registered to their service provider, this button light will turn on-off. User can press this button to try to register again to their service provider.
9	MESSAGE	This is for future use of functions upgrade.
10	Menu / OK	To bring out the menu selection while IP phone is in idle state. To be used as “OK” key while editing. To be used as “Dial Out” key when on dial mode.
11	Cancel	To escape to an upper layer menu selection or to cancel the current edit.
12	TXT ↔ NUM	Press this button and alphabets shown on the keypads will be displayed.
13	TRANSFER	To transfer an active call (incoming call answered or outgoing call accepted) to another IP phone.
14	Redial / Dial	Press to dial the last dialed number when the IP phone is off-hooked.
15	Speed Dial M1 – M10	To make a speed dial call by pressing the speed dial key M1 – M10.
16	Phone Book	By pressing this button to store up to 50 phone numbers.

1.1.2 Rear View



Rear Panel of VIP-103PT/VIP-153PT/VIP-153T

- | | | |
|---|--------------|---|
| 1 | PC | RJ-45 connector, to maintain the existing network structure, connected directly to the PC through straight CAT-5 cable |
| 2 | WAN | RJ-45 connector, for Internet access, connected directly to Switch/Hub through straight CAT-5 cable.
802.3af PoE support in PT model |
| 3 | 7V DC | 7V DC Power input outlet |

Hint

-
- The Power over Ethernet support on PLANET PoE IP phones complies with the 802.3af standards. Using non-802.3af compliant PoE device will burn up the IP Phone permanently.
-

Chapter 2

Configuring the PoE IP Phone through LCD Phone menus

2.1 Initialize VIP103PT/VIP-153PT/VIP-153T

STEP 1:

Power on machine, the LCD screen displays flashing greeting as below:

A rectangular box representing the LCD screen. Inside, the word "Starting" is displayed at the top, and three dots "..." are displayed below it.

STEP 2:

Wait for 10 seconds, the main LCD screen shall be shown as below, and when the Internet access is available, the VIP-103PT / VIP-153PT / VIP-153T will connect to the SNTP server. The current time will be displayed on the LCD.

A rectangular box representing the LCD screen. Inside, the text "Hi (GK off)" is displayed on the left side, and "Date Time" is displayed on the right side.

LCD screen of VIP-103PT (H.323)

A rectangular box representing the LCD screen. Inside, the text "SIP (Proxy off)" is displayed on the left side, and "Date Time" is displayed on the right side.

LCD screen of VIP-153PT/VIP153T (SIP)

STEP 3:

Press **MENU** to enter configuration mode then press **OK** button to enter sub menus; press **CANCEL** can jump out current menu to previous level.

1. **View**
2. **Configure (Password: 123)**
3. **Advanced Setting (Password: 1234)**

2.2 LCD Menu Configuration

LCD menu configuration for VIP-103PT

1st Level	2nd Level	3rd Level
IP PBX mode	1.1 (On/Off) If Yes: Name Number	
	1.2 Hotline	1.2.1 (Yes/No) Auto 1.2.2 (Yes/No) Manual If Yes: IP Address
IP Address	Check the active network parameters in machine	
View	2.1 Network Value	2.1.1 IP Address 2.1.2 Network Mask 2.1.3 Default Route 2.1.4 DNS Server
	2.2 IP PBX Mode	
	2.2 Ping	
	2.3 GK IP Address	
	2.4 Firmware Version	
Configure	Password: 123	
	3.1 Network	3.1.1 (Yes/No) Dynamic IP 3.1.2 (Yes/No) PPPoE If Yes: PPPoE User Name PPPoE Password 3.1.3 Fixed IP IP Address Network Mask Default Route DNS Server 3.1.4 GMT
	3.2 VOIP	3.2.1 Telephone Number 3.2.2 Password 3.2.3 H.323 ID 3.2.4 (Yes/No) GK mode 3.2.5 GK IP 3.2.6 RAS Port 3.2.7 Q931 Port 3.2.8 Frame Size (True/False) LowRate
	3.3 Forward Mode	3.3.1 (Yes/No) Immediate If Yes: Immediate Number 3.3.2 (Yes/No) Busy If Yes: Busy Number 3.3.3 (Yes/No) No Answer If Yes: No Answer Number No Answer Time
Reboot	Reset the PoE IP Phone	

LCD menu configuration for VIP-153PT/VIP-153T

1st Level	2nd Level	3rd Level
View		
	1.1 Network Value	1.1.1 IP Address 1.1.2 Network Mask 1.1.3 Default Route 1.1.4 DNS Server
	1.2 Ping	
	1.3 Restart	
	1.4 Firmware Version	
	1.5 (Yes/No) PKT Trace	
Configure	Password: 123	
	2.1 Network	2.1.1 (Yes/No) Dynamic IP 2.1.2 (Yes/No) PPPoE If Yes: PPPoE User Name PPPoE Password 2.1.3 Fixed IP IP Address Network Mask Default Route DNS Server 2.1.4 GMT
	2.2 SIP	2.2.1 Username 2.2.2 Telephone Number 2.2.3 Password 2.2.4 (Yes/No) Proxy mode If Yes: Proxy Server Address Proxy Port 2.2.5 (Yes/No) Outbound Proxy If Yes: Outbound Proxy Address Outbound Proxy Port 2.2.6 SIP Domain Name 2.2.7 Frame Size (True/False) LowRate
	2.3 Forward Mode	2.3.1 (Yes/No) Immediate If Yes: Immediate Number 2.3.2 (Yes/No) Busy If Yes: Busy Number 2.3.3 (Yes/No) No Answer If Yes: No Answer Number No Answer Time

“View” menu

In the “View” main menu, there are submenus, users may check these items for machine information and also provide some simple functions to check the network status of the IP Phone.

1.1 View current network settings

By selecting “View” → “Network”, you could see the active network parameters in machine.

1.2 Ping

By selecting “View” → “Ping” and key in the IP address or domain name of another device, the IP Phone can verify the connection between remote device. The “ping” command also can be used to verify the Internet connection status via pinging an Internet node

1.3 Restart

By selecting the “View” → “Restart”, the PoE IP Phone will reboot itself immediately. This “warm restart” is different from the power down/up (cold restart).

Hint

-
- Restarting the IP phone will perform the following tasks:
 - Disconnect all the voice communications, and Internet connections
 - Registration toward the gatekeeper/proxy server will be torn down.
-

1.4 Display current firmware version

By selecting the “View” → “Firmware Version”, the users can view current software version in the PoE IP Phone.

1.5 PK Trace for signal monitoring

PLANET PoE IP Phones integrate self-analysis packet tracing algorithm, which is able to perform machine signaling monitoring task, By enabling the “View” → “PKT Trace” to (Yes), the LCD screen will display signaling messages received, and this can be used to monitor the signals and identify the machine running status.

Network Configuration

The first thing in using the PoE IP Phone is to set the network configuration to let the IP Phone connect to Internet. Depends on the network environment and phone models, please use the proper method to configure the IP Phone to connect to Internet.

2.1.1 Dynamic IP Method (DHCP Client)

When the IP phone installed in a DHCP server enabled network environment (for example, cable modem connection or corporation internal network). Please browse to machine

network configuration menu, and enable the DHCP client in **Dynamic IP** selection menu and the IP phone will obtain the IP parameters dynamically, and connect to Internet automatically.
Configuration steps:

Configure	Password : 123	Network	(Yes/No) Dynamic IP
------------------	-----------------------	----------------	----------------------------

2.1.2 PPPoE

Most of the broadband network environment provided now by ISP is the ADSL connection. In the ADSL environment, the IP Phone can directly connect to the ADSL modem by setting the PPPoE account (username and password) provided by ADSL service provider, the configuration steps are shown below: .

Configure	Password : 123	Network	(Yes/No) PPPoE
------------------	-----------------------	----------------	-----------------------

Please browse to machine network configuration menu, and enable the PPPoE service in machine, and setup the username/password fields for Internet access.

PPPoE Username – please input the user name of the account given by ADSL ISP.

PPPoE Password – please input the password of the account given by ADSL ISP.

2.1.3 Fixed IP

Under the “**Fixed IP**” submenu, please key in the IP address, network mask and default router settings provided by your ISP or private IP address.

Configure	Password : 123	Network	Static IP
------------------	-----------------------	----------------	------------------

Hint

Once all the network settings are completed, please restart your IP phone. After rebooting, please check Internet connection status via **View** -> **Ping**, then insert in a public, live IP address (for example, 168.95.1.1) to ping it. When the response is ok, then the network configurations are completed.

2.1.4 GMT

Via selecting the time zone in the GMT menu, VIP-103PT/VIP-153PT/VIP-153T is able to retrieve time information from timeserver on the net.

Configure	Password : 123	Network	GMT
------------------	-----------------------	----------------	------------

2.2 Registration to H.323 gatekeeper / SIP Proxy Server

After setting up the network parameters, the IP Phone is able to perform registration to the H.323 gatekeeper (VIP-103PT) or SIP Proxy server (VIP-153PT/VIP-153T) when you've been assigned the account from VoIP vendor/service provider.

2.2.1 Registration to a H.323 gatekeeper (VIP-103PT)

When VIP-103PT is installed in an H.323 GK enabled network, the VIP-103PT registers to the gatekeeper for voice communications. Following descriptions are parameters to complete the registration.

Configure	Password : 123	H.323
------------------	-----------------------	--------------

Telephone Number:

Telephone phone number used to register toward the gatekeeper.

Password:

Password used to register to the gatekeeper.

This password will be encapsulated in the H.235 message for authentication purpose. Not every gatekeeper demands this value for authentication; please skip this field if your ISP does not provide this information.

H.323 ID:

The H.323 ID will be used to register to the gatekeeper. Not every gatekeeper demands this value for authentication; please skip this field if your ISP does not provide this information.



Note

The telephone number, H.323 ID and password fields are application-oriented parameters. Please be sure to check with your service provider to get the proper information before setting up these fields.

GK mode (Yes/No):

Selecting **Yes** will enable machine in GK mode, and register to the gatekeeper. (Machine default mode is Peer-to-Peer (**P2P**))

If you'd like to change the GK settings, it is suggested to log in machine web interface, and browse to H.323 configuration menu to adjust related parameters. When VIP-103PT fails to register with the gatekeeper, the IP phone still able to establish voice calls toward other calling parties Peer-to-Peer via dialing the IP address directly on the keypad.

2.2.1 Registration to a SIP Proxy Server (VIP-153PT/VIP-153T)

When the VoIP vendor/operator is running the SIP system, use the VIP-153PT/VIP-153T to register to the proxy server. Configure the following parameters to do the registration.

Configure	Password : 123	SIP
------------------	-----------------------	------------

Username:

The username used to register to Proxy server. If Proxy server doesn't request specific username, please enter the assigned telephone number here.

Telephone Number:

Please input the phone number to register to the proxy server.

Password:

Please input the password to register to the proxy server.

This password is encapsulated in the SIP Proxy-Authorization field using MD5 digest method for authentication purpose. Not every proxy server needs this field, if not needed, keep it empty.

Note

- The SIP telephone number, login ID and password fields are application-oriented parameters. Please be sure to check with your service provider to get the proper information before inputting parameters.
- VIP-153PT/VIP-153T follows the RFC-2617 in the SIP authentication applications.

Proxy mode (Yes/No):

Please check the application demands and select the appropriate operation mode. Selecting **Yes** to enable SIP proxy mode, and requires additional data input (such as the username, password, server information) to establish the voice communication, SIP proxy is mostly adopted in ITSP (Internet Telephony Service Provider). Selecting **No** will enable machine in **Peer-to-Peer** mode.

After the SIP proxy mode is enabled, two configuration menus will appear: one is **proxy server address**, to set the IP address or domain name address of the proxy server. Another is the **proxy server port**, to set the port of the proxy server; Default proxy server port is **5060**. Please do not change the proxy server port number unless your service provider gives these parameters.

If the IP Phone does not register to any proxy server, it still can call to other IP phone by calling the IP address directly.

Outbound Proxy (Yes/No):

Please set this item to **Yes** if the registration needs to pass through the Outbound Proxy server.

Registration / Startup message

When the network and registration configurations are set, please restart the IP Phone, the LCD display on the IP phone will show one of the following message depends on whether the registration is ok or not.

When the VIP-103PT is registering to a H.323 gatekeeper successfully, the LCD screen will display the following message:

Registering (Number) Date Time
--

Means that the VIP-103PT is working OK and ready for outgoing/incoming calls. The number inside braces () is the IP phone 's number.

Hi (GK off) Date Time

Means that the VIP-103PT is working OK and ready for outgoing/incoming calls. But the "GK mode " flag is "No", i.e. the VIP-103PT not need to register. In this case, can call the IP address of other VIP-103PT directly.

When the VIP-153PT/VIP-153T is registering to a SIP proxy server successfully, the LCD screen will display the following message:

Registering (Number) Date Time
--

Means that the VIP-153PT/VIP-153T is working OK and ready for outgoing/incoming calls. The number inside braces () is the IP phone 's number.

SIP (Proxy Off) Date Time

Means that the VIP-103PT/VIP-153PT/VIP-153T is working OK and ready for outgoing/incoming calls. But the "Proxy On" flag is "No", i.e. the VIP-103PT/VIP-153PT/VIP-153T not needs to register. In this case, can call the IP address of other VIP-153PT/VIP-153T directly.

When the PoE IP Phone is failed in registering to a H.323 gatekeeper or SIP proxy server, the LCD screen will display the following message:

RegFail (Failed message)

Date Time

The failed message could be one of the followings:

Duplicate: means that the registering number is duplicated with others, or the PoE IP Phone's previous registration information is still kept in the gatekeeper/proxy server if not unregistered last time (this could happened when the PoE IP Phone is powered off instead of restarted from the menu item). If the previous registration information is not cleared, may need to wait about 4 minutes to let PoE IP Phone register successfully again.

Security: means that the account (username/password/H.323 ID) is not correct, please check your account again.

When the **RegFail** message displayed, the IP phone cannot make any calls now. But the menu selection and the onhook/offhook function can work OK.

When the PoE IP Phone is failed in registering to a H.323 gatekeeper or SIP

DHCPFail (Number)

Date Time

Means that the PoE IP phone is configured to use DHCP to get IP address, but the DHCP procedure failed (can not find a DHCP server or the DHCP server rejected the IP assignment). The IP phone cannot make any calls and the onhook/offhook functions cannot work cannot hear any sound on the handset and the speaker. Only the menu selection can work, now.

The PoE IP phone will continuously try to get the IP address from the DHCP server,

so, if the DHCP server is responding, the **DHCPFail()** display will change to Hi() display to notify that the PoE IP phone is working OK.

When the PoE IP Phone is set to use the PPPoE method for network connection, but have some problem in finding the PPPoE server (ADSL modem), the LCD screen will display the following message:

PPPoE FindFail (Number) Date Time

Means that the PoE IP phone is configured to use PPPoE to get IP address, but the PPPoE procedure failed (can not find a PPPoE server or the network connection has some problem). The IP phone cannot make any calls, but the menu selection and the onhook/offhook function can work OK.

When the PoE IP Phone is set to use the PPPoE method for network connection, but have some problem in the PPPoE account, the LCD screen will display the following message:

PPPoE AuthFail (Number) Date Time

Means that the PoE IP phone is configured to use PPPoE to get IP address, but the PPPoE server refuse the connection because the username/password is not correct. The PoE IP phone cannot make any calls, but the menu selection and the onhook/offhook function can work OK.

When the PoE IP Phone is enable the “**Immediate Forward**”, the LCD screen will display the following message:

FWD (Number) Date Time

Means that the VIP-103PT is working OK and ready for outgoing/incoming calls. But for any incoming call, it will be forwarded to the “Immediate Forward Number”;

this PoE IP Phone will not ring.

2.3 Call Forward Configurations

The PoE IP Phone supports three different kinds of call forward functions, please select "**Configure**" "**Forward Mode**" for these three kinds of selection:

2.3.1 Immediate Forward

Under "**Forward Mode**" submenu, users are able to setup for immediate forward by selecting Yes on (Yes/No) Immediate and input the number he/she would like the call to be forwarded to. For example, when A calls B, and B has enabled the immediate forward with number 555-5555 on the IP phone. In this circumstance, this phone call from A will be forwarded to number 555-5555. When the "immediate forward" is enabled, the "busy forward" and "no answer forward" cannot be enabled at the same time.

2.3.2 Busy Forward

Under "**Forward mode**" submenu, users are able to setup for busy forward by selecting **Yes** on "(Yes/No) **Busy**" and input the number he/she would like the call to be forwarded to when this phone is busy (active). For example, when A calls B, and IP phone B is busy and "**busy forward**" has been enabled with number 555-5555, then, this phone call from A will be forwarded to number 555-5555.

2.3.3 No Answer Forward

Under "**Forward mode**" submenu, users are able to setup for no-answer forward by selecting Yes on "(Yes/No) No Answer" and input the number he/she would like the call to be forwarded to when this phone is not answered for the specified seconds. For example, when A calls B, and B's phone is not answered and has Yes on "(Yes/No) No Answer" with number 555-5555 and no-answer time equals 10, then, this phone call from A will be forwarded to number 555-5555 after ringing for 10 seconds no answered.

Chapter 3

Configuring the VIP-103PT/VIP-153PT/ VIP-153T through Web Pages

VIP-103PT/VIP-153PT/VIP-153T are equipped with a web-based graphical user interface that can cover most configurations and machine status monitoring. Via standard, web browser, you can configure and check machine status from anywhere around the world.

Step 1. Browse the IP Address of your IP phone

After TCP/IP configurations on the phone, you may now open your web browser, and input machine default IP address **192.168.0.1** to logon VIP-103PT/VIP-153PT/VIP-153T web configuration page.

Step 2. Input the login name and password

- Login name: **root**
- Password: Null (no password)

SIP Phone Web Management



■ Username:

■ Password:

Step 3. Enter the web interface main screen

After logging in machine, machine main screen is shown below.

1. Overview

Network, and firmware information of the PoE IP Phone will be displayed on machine homepage; more machine configuration menu can be found in the left frame:



- **Firmware version:** machine firmware version information
- **Network value:**
 - IP Address: 192.168.0.1
 - Subnet Mask: 255.255.255.0
 - Default routing gateway: 192.168.0.254
 - DNS Server: 168.95.1.1

Step 4. Start machine configurations

Most important items are **Network Configuration** and **H.323/SIP Configuration**. Please remember to configure these commands before start to work with VIP-103PT/VIP-153PT/VIP-153T.

iNote

After parameter modifications, it is required to browse to the "**System Configuration**" page to reboot VIP-103PT/VIP-153PT/VIP-153T to make changes effective.

2. Network Configuration

Please select the "**Network Configuration**" icon on the left menu; the following page will display to allow you to set all the network related configurations

Network configurations:

- **Dynamic IP:** enable/disable DHCP Client function.
- **PPPoE:** enable/disable PPPoE function.
 1. PPPoE Username: set PPPoE authentication User Name.
 2. PPPoE Password: set PPPoE authentication password.
- **Fixed IP**
 3. IP Address: IP Address of VIP-103PT/VIP-153PT/VIP-153T
 4. Subnet Mask: setup the subnet mask of VIP-103PT/VIP-153PT/VIP-153T
 5. Default routing gateway: setup default gateway of VIP-103PT/VIP-153PT/VIP-153T
 6. DNS Server: setup domain name server IP address.

Others:

- **MAC Address:** MAC address of the PoE IP Phone
- **SNTP Server Address:** setup SNTP Server Address
- **GMT:** setup time zone for SNTP Server time

① Note

-
- When SNTP server is available, enable VIP-103PT/VIP-153PT/VIP-153PT SNTP function to point to SNTP server IP address, so that VIP-103PT/VIP-153PT/VIP-153PT can obtain proper time information from the SNTP server.
 - You may enable different time zone to meet the location of VIP-103PT/VIP-153PT/VIP-153T. For example, the time zone in Taiwan should be set as 8, which mean GMT+8.
-

After setting up network parameters, be sure to apply settings (via **"DONE"** button) and browse to **"System Configuration"** menu to reboot the machine to make the settings effective

Connection Type	Data required.
Dynamic IP (DHCP client)	In most circumstances, it is no need to configure the DHCP settings.
PPPoE	The ISP will assign PPPoE username / password for Internet access,
Fixed IP Address	The ISP will assign IP Address, and related information.

i Hint

When there is difficulty connecting your IP phone to the Internet, please consult your ISP personnel to obtain proper PPPoE/IP address related information, and input carefully. If Internet connection cannot be established, please check the physical connection or contact the ISP service staff for support information.

3. H.323 Configuration (VIP-103PT)

Please select the “H.323 Configuration” icon on the left menu; the following page will displayed for H.323 related configurations.

H323 Configuration

■ **H323 Parameters:**

1. Telephone Number: 100
2. Password:
3. H.323 ID:
4. GK mode:
 1. GK IP:
 2. RAS Port: 1719
 3. Q931 Port: 1720

DONE

■ **Forward Mode:**

1. Immediate:
2. Immediate Number:
3. Busy:
4. Busy Number:
5. No Answer:
6. No Answer Number:
7. No Answer Time: 0

DONE

■ **Advanced Setting:**

1. RTP Port (Starting): 10000
2. RTP Port (Ending): 10004
3. H.323 Fast Connect:
4. H.245 Mode: Tunneling Separate None
5. H.235 Security: Common CAT SHA1
6. H.450.1 On:
Available packet time table

G.723 : 30ms / 60ms
G.729 : 20ms / 40ms
G.711u : 20ms
G.711a : 20ms
7. Codec: G.723 G.729 G.711u G.711a
8. Frame Size: 30ms 60ms
9. Full RRQ:

DONE

H.323 Parameters:

- **Telephone Number:** insert telephone number of VIP-103PT when registering to Gatekeeper.
- **Password:** the authorized password in the gatekeeper application
- **H.323 ID:** setup H.323 ID provided by service provider, the H.323 ID is mostly used as username for authentication when the IP phone is enabled in GK mode.

- **GK mode:** Enable/disable Gatekeeper function.
- **GK IP:** Set Gatekeeper IP address or Domain Name.
- **RAS/Q931 Port:** communication port number of the Gatekeeper for VIP-103PT to send messages.

Forward Mode:

- **Immediate:** enable/disable **Immediate Forward** function.
- **Immediate Number:** setup the immediate forward number here.
- **Busy:** enable /disable **Busy Forward** function.
- **Busy Number:** setup the busy forward number here
- **No Answer:** enable /disable **No Answer Forward** function.
- **No Answer Number:** setup the No Answer Forward number here
- **No Answer Time:** setup the No Answer Forward number here

Advanced Setting:

- **RTP Port (Starting):** the starting port number used in the PoE IP phone voice transmission port.
- **RTP Prot (Ending):** the ending port number used in the PoE IP phone voice transmission port.
- **H.323 Fast Connect:** enable/disable the H.323 fast start mode
- **H245 Mode:** enable H.245 tunneling mode or open the H.245 channel after H.323 fast start connection
- **H235 Security:** select the authentication type for the H.235 security
- **H.450.1 On:** enable or disable the H.323 supplementary services in machine
- **Codec (G.711u/G.711a/G.729/G.723):** voice codec used to encode the voice packets. In VoIP communications, the called party determines the type of voice codec used in conversation.
- **Frame Size:** user can set different packet size to meet the demands from calling party.
- **Full RRQ:** in the GK application, you may enable or disable sending full RRQ information when registering to the GK (Default: **No**)

3. SIP Configuration (VIP-153PT/VIP-153T)

Please select the “**SIP Configuration**” icon on the left menu; the following page will display to allow you to set all the SIP related configurations.

The screenshot shows a web interface titled "SIP Configuration" with three main sections:

- SIP Parameters:** A list of 13 numbered items with input fields and checkboxes:
 1. Username: [input field]
 2. Telephone Number: 100 [input field]
 3. Password: [input field]
 4. Proxy mode:
 5. Proxy Server Address: [input field]
 6. Proxy Port: 5060 [input field]
 7. Outbound Proxy:
 8. Outbound Proxy Address: [input field]
 9. Outbound Proxy Port: [input field]
 10. SIP Domain Name: [input field]
 11. STUN Server:
 12. STUN Address: [input field]
 13. STUN Server Port: [input field]A "DONE" button is located at the bottom right of this section.
- Forward Mode:** A list of 7 numbered items with input fields and checkboxes:
 1. Immediate:
 2. Immediate Number: [input field]
 3. Busy:
 4. Busy Number: [input field]
 5. No Answer:
 6. No Answer Number: [input field]
 7. No Answer Time: 0 [input field]A "DONE" button is located at the bottom right of this section.
- Advanced Setting:** A list of 4 numbered items with input fields and radio buttons:
 1. Reg Expire: 60 [input field]
 2. RTP Port: 10000 [input field]
 3. Codec: G.711u G.711a G.729 G.723
 4. Frame Size: 30ms 60msA "DONE" button is located at the bottom right of this section.

SIP Parameters:

- **Username:** the username to register to Proxy Server. If Proxy server doesn't request specific username, please enter line number here.
- **Telephone Number:** the telephone number of VIP-153PT/VIP-153T to register to SIP proxy.
- **Password:** the password used to register to SIP proxy. This parameter will not be necessary when SIP proxy doesn't request client to set password. In the

circumstances, please setup the username same as telephone number for SIP registration.

- **Proxy mode:** enable/disable SIP proxy mode.
- **Proxy Server Address:** SIP proxy IP address or domain name.
- **Proxy Port:** setup proxy port for SIP proxy communication.
- **Outbound Proxy:** Enable/disable SIP outbound proxy functions.
- **Outbound Proxy Address:** IP address / domain name of the SIP outbound proxy.
- **Outbound Proxy Port:** communication port number of the SIP outbound proxy
- **SIP Domain Name:** domain name.of the SIP STUN server.
- **STUN Server:** Enable/disable SIP STUN function.
- **STUN Address:** IP address / domain name.of the SIP STUN server.
- **STUN server Port:** communication port number of the SIP STUN server.

Forward Mode:

- **Immediate:** enable/disable immediate call forwarding.
- **Immediate Number:** the immediate call forwarding number.
- **Busy:** enable/disable busy call forwarding.
- **Busy Number:** the busy call forwarding number.
- **No Answer:** enable/disable no-answer call forwarding in machine.
- **No Answer Number:** the no-answer call forwarding number.
- **No Answer Time:** the waiting time to forward the call to specific number.

Note

-
- Settings in the SIP configuration menu are mostly application-oriented. Please be sure to check with the service provider to obtain proper information before setting up machine
 - Please contact the service provider or VoIP system administrator to have the proper call forwarding settings/information.
-

Advanced Setting:

- **Reg Expire:** modify the value "Expires:" field in the SIP/SDP messages. This will control the re-registration period.
- **RTP Port:** the port number used in the PoE IP phone voice transmission port.
- **Codec (G.711u/G.711a/G.729/G.723):** voice codec used to encode the voice packets. In SIP communications, the called party determines the type of voice codec used in conversation.
- **Frame Size:** user can set different packet size for each codec.

4. Phone Configuration (VIP-103PT)

Please browse to the “**Phone configuration**” on the left menu, following page will be displayed for phone call and voice codec related configurations.

Phone Configuration

■ **IP PBX Setting:**

1. On/Off:
2. Name:
3. Number:

■ **Hot Line Setting:**

1. Auto:
2. Manual:
3. IP Address:

■ **Net Action:**

1. Net Setting: None Transfer Conference

■ **Prefix Setting:**

1. International Code:
2. Country Code:
3. Area Prefix Code:
4. Message Number:

■ **Voice:**

1. Ring Volume(0-9):
2. Handset Volume(0-9):
3. Handfree Volume(0-9):
4. CodecTxGain(0-9):

IP PBX Setting:

- **IP PBX Setting:** enable/disable IP PBX support.
- **Name:** extension name displayed on the LCD screen
- **Number:** extension number displayed on the LCD screen

 Note

- Current supported IP PBX system: PLANET IPX-1000/IPX-1500
-

Hot Line:

Auto/Manual: please set up the auto/manual function to enable/disable hot line function.

IP address: if the hot line function is use manual mode, please input the IP address of the destination device.

4. Phone Configuration (VIP-153PT/VIP-153T)

Please browse to the “**Phone configuration**” on the left menu, following page will be displayed for phone call and voice codec related configurations.

The screenshot shows a web interface titled "Phone Configuration" with a grey background. It contains five sections, each with a "DONE" button:

- IP PBX Setting:** 1. IP PBX Setting: DONE
- Prefix Setting:** 1. International Code:
2. Country Code:
3. Area Prefix Code: DONE
- Voice:** 1. Ring Volume(0-9):
2. Handset Volume(0-9):
3. Handfree Volume(0-9):
4. CodecTxGain(0-9): DONE
- Hot Line:**
1. Dest URL:
2. Tel Number: DONE
- Others:** 1. VAD:
2. BG Noise Level(0-9): DONE

IP PBX Setting:

- **IP PBX Setting:** enable/disable IP PBX function.

Prefix Setting:

This parameter is used to setup the calling prefix string.

For some specific VoIP system, the IP Phone can call to PSTN/mobile number and the called number must include the country code, e.g. the country code of Taiwan is 886, so if want to call a Taipei city call 02-12345678, the called number needed to be 886-2-12345678. But this is not consistent with the normal dialing behavior. In order to use the normal dialing behavior, you can set the “My Country Code” value to 886, and

set the "Area Prefix Code" value to 0.

As for the "International Code", when you want to dial an international call, the International Code will be removed. E.g., suppose want to make a call to China, the dialed number will be 002-86-2112341234, the IP Phone will remove the "International Code" 002, and send the real number of 86-2112341234.

This prefix code implementation is only needed for some specific VoIP system.

Normally, this is not needed; just let all the prefix values be empty.

Voice:

- **Ring Volume (0-9):** select the volume of ringing.
- **Handset Volume (0-9):** select the telephone set volume
- **Handfree Volume (0-9):** select the volume of handfree speaking.
- **CodecTxGain(0-9):** select the output voice volume.
-

Hot Line:

Hot Line: enable/disable hot line function.

Destination URL: please input the IP address of the destination device.

Telephone Number: please input the telephone number of the destination device.

Others:

- **VAD:** VAD is short for Voice Activity Detection, the IP Phone will detect if the user is talking or not, and avoid to send voice packet when the user is silent to decrease the bandwidth requirement.
- **BG Noise Level (0-9):** signal level of back ground noise. This parameter controls the sensitivity of the VAD detection.

5. System Configuration

Please select the "**System Configuration**" on the left menu; you may set up the system related configurations.

In this page, you can perform machine software upgrade, change username/password of the web login, and reboot (restart) machine.

System Configuration

- **Firmware Upgrade:**
 1. FTP Server:
 2. Image File Name:
- **Change web Username & Password:**
 1. Username:
 2. Password:
 3. Confirm New Password:
-

Firmware Upgrade:

- **FTP Server:** IP address of the FTP server, which contains the firmware file of the PoE IP phone
- **Image File Name:** firmware file name of VIP-103PT/VIP-153PT/MIP-153T.

Change web Username & Password: The default username/password for web login is **root** / **<no password>**, you may modify the username/password for more security. Please enter the new login name, re-enter the current password, new password and confirm new password to enable new password.

Reboot Button: Press the **"REBOOT"** button will reset the PoE IP Phone

Hint

-
- The PoE IP phones are optimized for VoIP applications, it will not require firmware upgrade in most cases. It is suggested to check with our local distributor for machine firmware upgrade information.
 - While performing firmware upgrade, please allocate the firmware file in the FTP root directory, so that the IP phone can find the firmware file to complete the upgrade process.
 - When pressing the **"REBOOT"** button, all the modified data will automatically be saved in machine, and activated in the next power-on.
 - If you'd like to load the factory default value into machine, please press **"#*110*"** on the keypad to reset the IP phone.
-

6. Speed Dial

Please select the “Speed Dial” on the left menu; users may set up the speed dial related configurations.

- **M1~M10 Name:** user can specify 10 sets of speed dial via web interface. Please input name of the destination device
- **M1~M10 Number:** please input IP address and URL of the destination device

7. Phone Book

Please select the “Phone Book” on the left menu; users may set up the phone book related configurations.

- **Add Entry:** user can specify 50 sets of phone book via web interface. Please input index, name and IP address/URL of the destination device
- **Delete Entry:** user can delete any configured phone book data by index.

Chapter 4

Voice communication samples

There are several ways to make calls to desired destination in VIP-103PT/VIP-153PT/VIP-153T. In this chapter, we'll lead you step by step to establish your first voice communication via web browsers operations.

Peer to peer (P2P) mode

VIP-103PT to VIP-103PT connection via IP address

Assuming there are two VIP-103PT in the network the IP address are 192.168.0.1, 192.168.0.2



In **peer-to-peer** (P2P) mode, you can direct pickup the handset and dial the IP address of the remote VIP-103PT using the keypad as below:

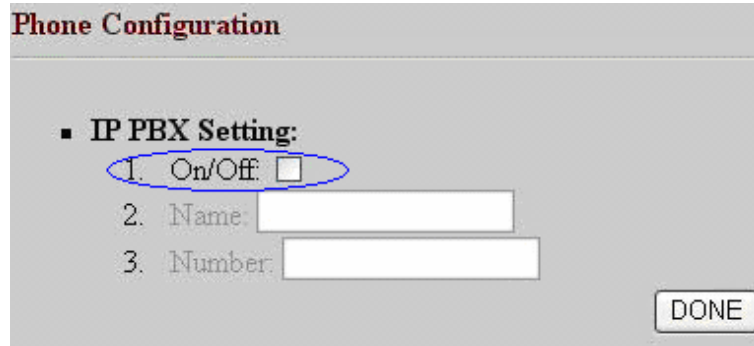
1 9 2 * 1 6 8 * 0 * 2 #

Then the phone of 192.168.0.2 should ring. You can do the same thing to the second VIP-103PT.

IP PBX extension configurations – IPX-1000/IPX-1500

Machine configurations on VIP-103PT

- a) Log in machine via web browser, and browse to the H.323 configuration page, and enable the "IPX " feature.



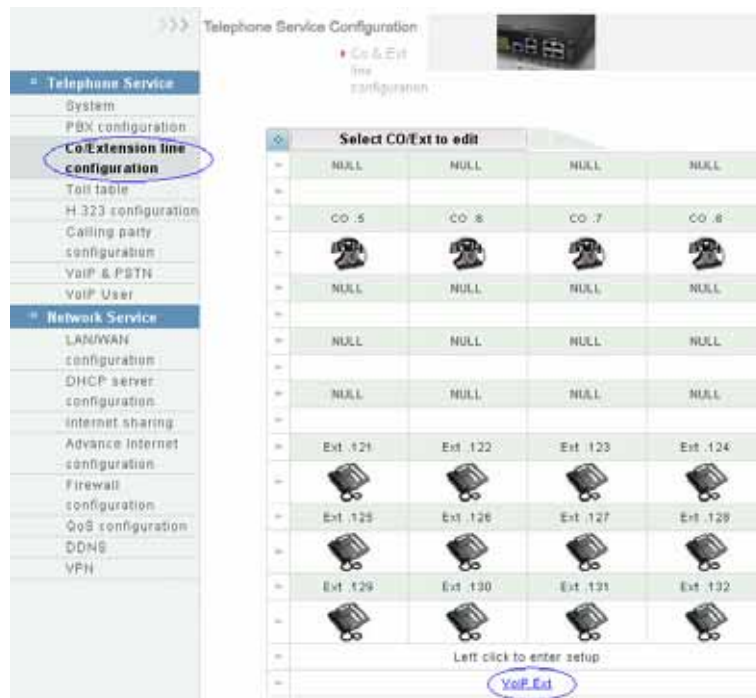
- b) After enabling the "IP extension" feature, please enable hotline feature, and insert the IP address of IP PBX system. PLANET IPX-1000, IPX-1500).

After these configurations, be sure to click the "DONE" button to apply settings and browse to "System Configuration" menu to reboot the machine to make the settings effective.

Note: if the VIP-103PT setting DHCP client to connect the IPX-1500 DHCP server, in the hot list setting, please set up auto mode to connect the IPX-1500.

Machine configurations on IPX-1500

- a) Log in machine, and browse to the Extension configuration page. Select the "VoIP Ext." to start the IP extension related settings.



- b) In the VoIP extension configuration page, please insert the IP address, MAC address information of the VIP-103PT.

Add VoIP Ext		
▶ Ext # No	<input type="text"/>	Ext phone number
▶ Phy line	35 ▼	Physical line number
▶ IP	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>	VoIP Ext IP
▶ Mac	<input type="text"/> - <input type="text"/> - <input type="text"/> - <input type="text"/> - <input type="text"/> - <input type="text"/>	VoIP Ext MAC
▶ Password	1234	Ext user access password

Assign and save the extension related parameters for the VIP-103PT.

Add VoIP Ext		
▶ Ext # No	<input type="text"/>	Ext phone number
▶ Phy line	35 ▼	Physical line number
▶ IP	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>	VoIP Ext IP
▶ Mac	<input type="text"/> - <input type="text"/> - <input type="text"/> - <input type="text"/> - <input type="text"/> - <input type="text"/>	VoIP Ext MAC
▶ Password	1234	Ext user access password
▶ Answer group	01 ▼ 00 ▼ 00 ▼	Answer CO call group
▶ Dial out group	01 ▼ 00 ▼ 00 ▼	Use CO group
▶ Out call ctrl mode	LockExt ▼	Ext phone long distance call control mode
▶ Out call priority	Unlimit ▼	Ext long distance call priority
▶ CO to COTx auth.	Disable ▼	Enable transfer CO call follow to another CO line
▶ COTx to Ext	Allow CO DirectCall ▼	Disallow CO direct call to ext user, only accept CO transfer by other ext
▶ Talk time warning	Disable ▼	Enable warning when ext connect CO too long
▶ Ext direct to CO	Disable ▼	Define the ext pickup to CO mode
▶ Voice mail ctrl	Disable ▼	Ext user voice mail function on/off
▶ Voice mail to mail ctrl	Disable ▼	Ext user voice mail To mail function on/off
		DONE
		Back To List

- c) In the VoIP extension configuration page, please insert the IP address, MAC address information of the VIP-103PT.

After these settings, the VIP-103PT is able to act as one of the IP extensions under the IPX-1000, IPX-1500.

Peer to peer (P2P) mode

VIP-103PT to VIP-103PT connection via IP address

Assuming there are two VIP-103PT in the network the IP address are 192.168.0.1, 192.168.0.2



In **peer-to-peer** (P2P) mode, you can direct pickup the handset and dial the IP address of the remote VIP-103PT using the keypad as below:

1 9 2 * 1 6 8 * 0 * 2 #

Then the phone of 192.168.0.2 should ring. You can do the same thing to the second VIP-103PT.

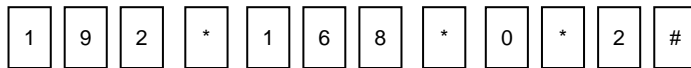
VIP-153PT to VIP-153T connection via IP address

Assume there are two SIP IP phone in the network the IP address are 192.168.0.1, 192.168.0.2





In **peer-to-peer (P2P)** mode, you can direct pickup the handset and dial the number using the keypad as below:



Then the phone of 192.168.0.2 should ring. You can do the same thing to the second VIP-153T.

i Hint

-
- If the IP address of the remote calling party is known, you may directly make calls via its IP address and end with an "#".
 - If the IP phones are installed behind a NAT/firewall/IP sharing device, please make sure the NAT device support SIP applications before making calls
-

VIP-153PT to VIP-153T connection via SIP telephone number



STEP 1:

Please browse machine **"SIP configuration"** menu, and enable the **Proxy mode**. Insert IP address of the remote calling party in the **"Proxy Server Address"** field. Sample configuration screen is shown below:

■ **SIP Parameters:**

1. Display Name: 100
2. Login ID:
3. Telephone Number: 100
4. Password:
5. Proxy mode:
6. Proxy Server Address: 192.168.0.2
7. Proxy Port: 5060
8. Outbound Proxy:
9. Outbound Proxy Address:
10. Outbound Proxy Port:

STEP 2:

After these configurations, be sure to click the “**DONE**” button to apply settings and browse to “**System Configuration**” menu to reboot the machine to make the settings effective. After rebooting, the unit will communicate with the remote calling party, the LCD screen will show below:

Registering (100)
Nov 16 12:10

STEP 3:

At this moment, you may pick up the handset and dial “200” to connect the remote VIP-153T to start the voice communications.

Voice communication via SIP proxy server – SIP-50



■ VIP-153PT WAN IP Address: 192.168.0.1
Line Number: 1001

■ VIP-153T WAN IP Address: 192.168.0.2
Line Number: 2002

Machine configurations on the VIP-153PT/VIP-153T:

STEP 1:

Log in SIP-50 and create two testing accounts/password: **1001/123** (for VIP-153PT), and **1002/123**(for VIP-153T) for the voice calls.

STEP 2:

Please log in VIP-153PT via web browser, browse to the **SIP setting** menu and select the **Domain Service** config menu. In the setting page, please insert the account/password information obtained from your service provider (in this sample, we're using PLANET SIP-50 as the SIP Proxy server for SIP account, call authentications), and then the sample configuration screen is shown below:

■ SIP Parameters:

1. Username: planet
2. Telephone Number: 1001
3. Password: ●●●
4. Proxy mode:
5. Proxy Server Address: 192.168.0.50
6. Proxy Port: 5060
7. Outbound Proxy:
8. Outbound Proxy Address:
9. Outbound Proxy Port:
10. SIP Domain Name:
11. STUN Server:
12. STUN Address:
13. STUN Server Port:

DONE

STEP 3:

Repeat the same configuration steps on VIP-153T, and check the machine registration status, make sure the registrations are completed.

Test the scenario:

To verify the VoIP communication, you may make calls from SIP client (VIP-153PT) 1001 to the number 1002 (VIP-153T) or reversely make calls from SIP client (VIP-153T) 1002 to the number 1001 (VIP-153PT)

Voice communication via IP PBX system – IPX-2000 (Auto-config)

In the following sample, we'll introduce how to integrate the SIP Phone with our IP PBX system IPX-2000 via Auto-config feature.



Machine configurations on the IPX-2000:

STEP 1:

Log in IPX-2000 and browse to the **DHCP** menu and create new options list for the auto configuration.

DHCP

Enable Disable Save Cancel changes Delete Show client

DHCP Pool

<Add new >

lan

ID: lan

Single host :

Range: 192.168.1.101 ~ 192.168.1.200

Options list:

lan,150,192.168.1.1 Delete

Code	Value
151	http://192.168.0.50/tftpboot Add

Code: please insert **151** as the DHCP server option.

Value: http://LAN IP for IPX-2000/tftpboot

If you'd like to enable auto-config for IP extension features in IPX-2000, please be sure to setup the DHCP option code and the value information.

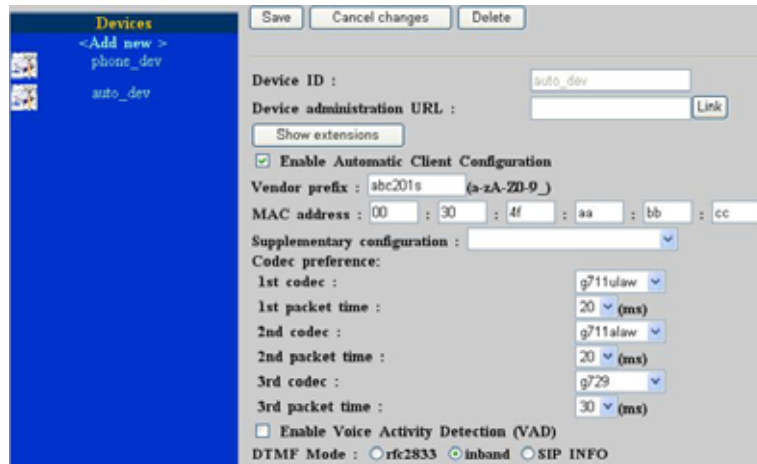
In most case, insert the optional code 151 and the value=http://192.168.0.50/tftpboot/

Note

- 192.168.0.50 is the IP address of IPX-2000

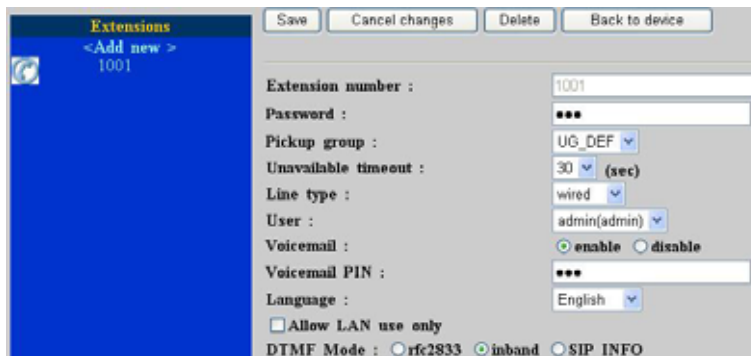
STEP 2:

Please browse to the **Device** menu and create new device for the auto configuration.



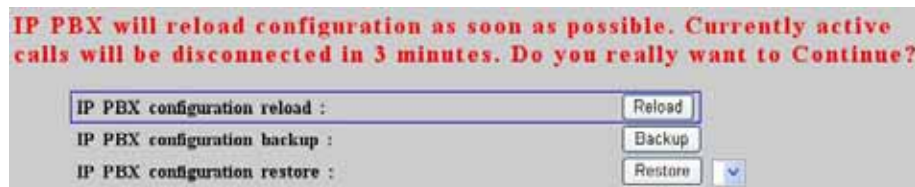
STEP 3:

Please press the **Show extensions** button to create the two extension accounts/password: **1001/123** (for VIP-153PT), and **1002/123**(for VIP-153T) for the voice calls.



STEP 4:

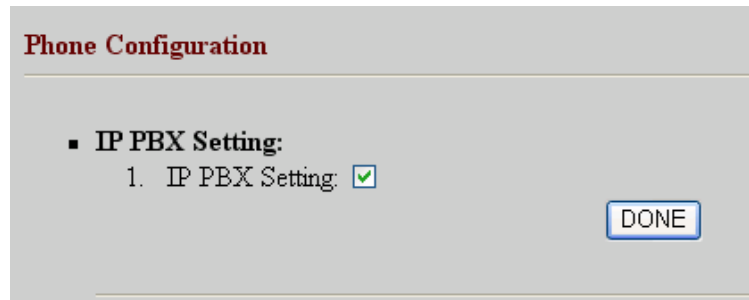
After setting up the parameters, please refer to the path to activate the settings :**Service ---> IP PBX service ---> IP PBX configuration reload**



Machine configurations on the VIP-153PT/VIP-153:

STEP 5:

Please log in VIP-153PT via web browser, please browse to the **Phone Configuration** page, and enable the IPX PBX setting features for IP PBX system. After these configurations, be sure to click the **“DONE”** button to apply settings and browse to **“System Configuration”** menu to reboot the machine to make the settings effective.



STEP 6:

After enabling the Auto-config feature, the VIP-153PT shall be able to obtain IP address and SIP extension information from IP PBX system IPX-2000 information. The VIP-153PT will perform registration to IPX-2000 after obtaining the extension config file.

STEP 7

Repeat the same configuration steps on VIP-153T, and check if the VIP-153T is successfully registered with the IPX-2000 as one of the IP extensions.

Test the scenario:

To verify the VoIP communication, you may make calls from extension side (VIP-153PT) 1001 to the number 1002 (VIP-153T) or reversely make calls from extension client (VIP-153T) 1002 to the number 1001 (VIP-153PT)

Appendix A

IP Phone Specifications

Product	H.323 PoE IP Phone	SIP IP Phone	SIP PoE IP Phone
Model	VIP-103PT	VIP-153T	VIP-153PT
Hardware			
WAN	1 x 10/100Mbps RJ-45 port Power Over Ethernet 802.3af compliant at PT model		
PC	1 x 10/100Mbps RJ-45 port		
LCD display	2 x 16 characters		
Speaker	8 Ohm/0.2 Watt speaker for speakerphone operation		
Protocols and Standard			
Standard	H.323 version 4,H.323 Fast start, H.245 Tunneling and H.245 DTMF relay	SIP 2.0 (RFC3261), SIP digest authentication (MD5),	
Voice codec	G.723.1 (6.3k/5.3k), G.729A/B, G.711 (a-law/u-law)		
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) Dynamic Jitter Buffer		
Supplementary services	Immediate (unconditional) call forwarding Busy call forwarding No answer calls forwarding Call hold/transferring.		
Call history	Record incoming call Outgoing call Missed (not accepted) call history		
Protocols	SIP 2.0 (RFC-3261), H.323, TCP//IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, FTP, PPP, PPPoE		
Network and Configuration			
Access Mode	Static IP, PPPoE, DHCP		
Management	Web, Keypad, auto-config by IPX-2000		
Dimension (W x D x H)	170 mm x 220 mm x 60 mm		
Operating Environment	0~40 degree C, 10~95% humidity		
Power Requirement	7V DC (802.3af 48VDC in line power at PT model)		
EMC/EMI	CE, FCC Class B		